REMARKS

The claims are claims 1 to 3, 6, 9 to 11, 14 and 17 to 22.

The application has been further amended at several locations to correct minor errors and to present uniform language throughout. A new ABSTRACT is proposed within the current word limit.

Claims 4, 5, 7, 8, 12, 13, 15 and 16 have been canceled. New claims 17 to 22 are added. New claims 17 to 22 recite calculation of the noise estimate as disclosed in the original application at page 10, lines 2 to 20.

Claims 1 to 16 were rejected under 35 U.S.C. 103(a) as made obvious by S. Bloebaum et al. U.S. Patent 6,070,137, filed January 7, 1998.

Claims 1 and 9 recite subject matter not made obvious by Bloebaum et al. Claim 1 recites "multiplying the samples by a windowing function." In a similar fashion claim 9 recites the noise suppression circuit is operable to "multiply the samples by a windowing function." The OFFICE ACTION suggests at page 3, line 6 that Bloebaum et al discloses the claimed windowing function in his disclosed framing. The Applicants dispute that the framing of Bloebaum et al makes obvious the claimed windowing function. Both claims 1 and 9 separately recite selecting "a fixed number of samples" as the framing function. The framing in Bloebaum et al fails to make obvious the multiplication required by claims 1 and 9.

The OFFICE ACTION states at page 4, lines 6 to 10:

> "As per claims 3 and 11, Bloebaum et al. do not say what inherent window they are using. However, an artisan at the time of invention would have known to use a Hanning (raised cosine) window because of its notoriously well-known convenience of enabling "unwindowing" by addition after inverse FFT when using 50 percent Hanning time window overlap."

While the Hanning window is known, the OFFICE ACTION provides no evidence that of use of this windowing in noise suppression. The Applicants respectfully submit that Bloebaum et al does not mention Hanning windowing and thus fails to make obvious use of this windowing in noise suppression. Accordingly, claims 1 and 9 are allowable over Bloebaum et al.

Claims 1 and 9 recite further subject matter not made obvious by Bloebaum et al. Claim 1 recites "selecting half of the transformed windowed signals." Claim 9 similarly recites the noise suppression circuit is operable to "select half of the transformed windowed signals." The OFFICE ACTION recites at page 3, lines 8 to 10:

> "using only half the Fourier-transformed data (single-sided) because of the complex-conjugate symmetry of a Fast Fourier Transform of real signals alluded to (col. 5, lines 8-10)"

Bloebaum et al states at column 5, lines 4 to 20 (including the lines cited by the Examiner):

> "The transformer function 32 includes a discrete Fourier transform (DFT) 42 which receives a frame of time-domain audio samples. The DFT 42 computes the complex spectrum $S(e^{j\omega})$ at K uniformly spaced discrete frequencies, $\omega=\pi i/K$, $0\leq i<K$. Note that a single-sided, frequency-domain representation is feasible given the complex symmetry produced by real-valued input signals such as audio. The DFT 42 is typically realized by a fast Fourier transform (FFT) algorithm which provides certain implementation advantages. The size of the DFT or FFT is dependent on the audio frame size. For example, a 160-sample audio frame may be transformed by a 256-point FFT, with ninety-six samples from the previous frame included. The output of the DFT 42 is input to block 44 which computes a power spectral density (PSD) estimate for the current frame, represented by $|S(e^{j\omega})|^2$. This PSD estimate is calculated at the same set of discrete frequencies as $S(e^{j\omega})$."

The Applicants respectfully submit that the Examiner has misapplied the teachings of Bloebaum et al. Bloebaum et al teaches employing single-sided frequency-domain representation during computation of the discrete Fourier transform via an FFT algorithm (see column 5, lines 8 to 10). This use of half of the data is in the production of the DFT/FFT data. Claims 1 and 9 recite that the fast Fourier transform yields "transformed windowed signals" and selecting half of these "transformed windowed signals." Bloebaum et al fails to make obvious selection of half the results of his DFT 42 as would be required by claims 1 and 9. Accordingly, claims 1 and 9 are allowable over Bloebaum et al.

Claims 1 and 9 recite further subject matter not made obvious by Bloebaum et al. Claim 1 recites "calculating a smoothed power estimate by smoothing the power estimate over time." Likewise, claim 9 recites the noise suppression circuit operates to "calculate a power estimate of the transformed windowed signals." The OFFICE ACTION states at page 3, lines 10 and 11:

> "smoothing the power estimate over time when there is no speech to calculate a noise power estimate (col. 5, lines 37-44 and 60-65)"

Bloebaum et al states at column 5, lines 30 to 44:

> "The adaptation process involves smoothing of the model parameters in order to reduce the variance of the noise estimate. This may be done using either a moving average (MA), autoregressive (AR), or a combination ARMA process. AR smoothing is the preferred technique, since it provides good smoothing for a low ordered filter. This reduces the memory storage requirements for the noise suppression algorithm. The noise model adaptation with first order AR smoothing is given by the following equation:
>
> $$N^{(i)} = \alpha N^{(i-1)} + (1-\alpha) S,$$

where α may be in the range $0 \leq \alpha \leq 1$, but is further constrained to the range $0.8 \leq \alpha \leq 0.95$ in the preferred embodiment of the invention."

This portion of Bloebaum et al clearly teaches smoothing of the noise estimate and not smoothing the power estimate as claimed. The Applicants respectfully submit that Bloebaum et al fails to teach smoothing of the power estimate. Bloebaum et al states at column 5, lines 60 to 62:

> "The Variance Reduction block receives as input $|S(e^{j\omega})|^2$ and applies a smoothing function in the frequency domain to generate an output $|S\hat{}(e^{j\omega})|^2$. "

The Applicants respectfully submit that this smoothing in the frequency domain cannot make obvious the smoothing "over time" recited in claims 1 and 9. Accordingly, claims 1 and 9 are allowable over Bloebaum et al.

Claims 1 and 9 recite further subject matter not made obvious by Bloebaum et al. Claim 1 recites "adding the sampled speech signal to a portion of the speech signal of a previous frame.". Claim 9 recites that the noise suppression circuit operates to "add the sampled speech signal to a portion of the speech signal of a previous frame." The Applicants respectfully submit that Bloebaum et al fails to make obvious this adding of the "sampled speech signal" and the "speech signal of a previous frame." Note that the OFFICE ACTION fails to point out where Bloebaum et al makes obvious this subject matter. In the absence of any allegation that this subject matter is obvious, claims 1 and 9 are allowable.

Claims 2, 3, 10 and 11 are allowable by dependency upon allowable base claims.

Claims 6 and 14 recite subject matter not made obvious by Bloebaum et al. Claim 6 recites "the noise estimation is calculated

by increasing a noise spectral estimate by a small margin." Claim 9 recites the noise suppression circuit operates wherein "the noise estimation is calculated by increasing a noise spectral estimate by a small margin." The Applicants respectfully submit that Bloebaum et al fails to make obvious this increase of the noise spectral estimate by "a small margin." Note that the OFFICE ACTION fails to point out where Bloebaum et al makes obvious this subject matter. In the absence of any allegation that this subject matter is obvious, claims 6 and 14 are allowable.

New claims 17 to 22 recite subject matter not made obvious by Bloebaum et al. New claims 17 and 20 recite limitations on the calculation of the noise estimate based upon predetermined constants upconst and downconst. This subject matter is not made obvious by Bloebaum et al. Claims 18 and 21 recite limitations on the constant upconst not made obvious by Bloebaum et al. Claims 19 and 22 recite limitations on the constant downconst not made obvious by Bloebaum et al. Accordingly, claims 17 to 22 are allowable.

The Applicants respectfully submit that all the present claims are allowable for the reasons set forth above. Therefore early reconsideration and advance to issue are respectfully requested.

If the Examiner has any questions or other correspondence regarding this application, Applicants request that the Examiner contact Applicants' attorney at the below listed telephone number and address to facilitate prosecution.

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VERSION WITH MARKINGS TO SHOW CHANGES MADE

Note inserted text is marked by underlining and deleted text is marked by ~~strikeout~~.

In the Abstract

Replace the ABSTRACT OF THE DISCLOSURE with the following:

--A system for reducing noise in an acoustical signal ~~is provided. The system~~ comprises a sampler (104) for obtaining discrete samples of the acoustical signal, an analog to digital converter (106) ~~coupled to the sampler (104) and operable to convert the analog discrete samples into a digitized sample~~, and a noise suppression circuit (108) ~~coupled to the analog to digital converter (106)~~. The noise suppression circuit (108) ~~reduces noise by first receiving the analog discrete samples and then selecting~~ selects a fixed number of samples. These samples are multiplied by a windowing function and the fast Fourier transform ~~of the windowed samples~~ is computed to yield transformed windowed signals. ~~Half of the transformed windowed signals are selected and a power estimate of the transformed windowed signals is calculated. Next, a~~ A smoothed power estimate ~~is calculated by smoothing the power estimate over time~~ and a noise estimate ~~is~~ are calculated. The noise estimate and the smoothed power estimate is used to calculate a gain function. A transformed speech signal is obtained by multiplying the gain function with the transformed windowed signal. Then, the inversed fast Fourier transform of the transformed speech signal is ~~calculated to yield a sampled speech signal and the sampled speech signal is~~ added to a portion of the speech signal of a previous frame.--

In the Specification

Rewrite the paragraph at page 4, lines 8 to 32 as follows:

--In one embodiment of the present invention a system for reducing noise in an acoustical signal is provided. The system comprises a sampler for obtaining discrete samples of the acoustical signal, an analog to digital converter coupled to the sampler and operable to convert the analog discrete samples into a digitized sample, and a noise suppression circuit coupled to the analog to digital converter. The noise suppression circuit reduces noise by first receiving the analog discrete samples and then selecting a fixed number of samples. These samples are multiplied by a windowing function and the fast Fourier transform of the windowed samples is computed to yield transformed windowed signals. Half of the transformed windowed signals are selected and a power estimate of the transformed windowed signals is calculated. Next, a smoothed power estimate is calculated by smoothing the power estimate over time and a noise estimate is calculated. The noise estimate and the smoothed power estimate ~~is~~ are used to calculate a gain function. A transformed speech signal is obtained by multiplying the gain function with the transformed windowed signal. Then, the inversed fast Fourier transform of the transformed speech signal is calculated to yield a sampled speech signal and the sampled speech signal is added to a portion of the speech signal of a previous frame.--

Rewrite the paragraph at page 7, line 19 to page 8, line 3 as follows:

--FIGURE 2 illustrates a block diagram illustrating noise suppression unit 108 in accordance with the teaching of the present invention. Illustrated is a frame buffer 200 coupled to a windowing unit 202 which is coupled to a fast Fourier transfer module 204 which is then coupled to a noise reduction algorithm unit 206 which is then coupled to a inverse fast Fourier transfer module 208 which is finally coupled to a noise suppression frame buffer 210. In

operation, frame buffer 200 partitions speech samples into frames of 32 ~~sample sizes~~ samples. The sample frames are then sent to the windowing module 202 or an appropriate window function is applied. In one embodiment a ~~hanning~~ Hanning window is applied. Fast Fourier transfer module 204 converts the frames to the frequency domain by using the well-known fast Fourier transform. Noise reduction unit 206 then invokes the main noise reduction algorithm. Noise reduction unit 206 takes the first 16 samples and computes the absolute value of the power of the sample. Then that power value is smoothed using the following equation.--

Rewrite the paragraph at page 9, lines 7 to 8 as follows:

--In step 302, the samples are multiplied by a ~~hanning~~ Hanning window. A ~~hanning~~ Hanning window is of the form--

Rewrite the paragraph at page 10, line 6 as follows:

-- then $n^n(i) = \text{downconst} * (n^{n-1}(i))$.--

Rewrite the paragraph at page 11, line 15 to page 12, line 2 as follows:

--In step 318, the inverse fast Fourier transfer is taken and in step 320, the sixteen computed values are added to the previous sixteen values. Then, in decision block 322 it is determined if there are any more already computed fast Fourier transition results awaiting calculation. If yes, the next 16 values are then calculated as before starting at step 308. If there are no more already calculated fast Fourier transfer value, decision box 324 is reached. In that box, it is determined it there is any more samples to ~~evolve~~ solve. If no, then the method ends at step 326. If there are more samples, execution continues at step 300.--

In the Claims

Please amend the claims as follows:

Cancel claims 4, 5, 7, 8, 12, 13, 15 and 16.

Add new claims 17 to 22 as follows:

17. (New) The method of claim 1, wherein:
said step of calculating a noise estimate includes

if a current smoothed power estimate is greater than a product of a predetermined constant upconst and a prior noise estimate, then setting a current noise estimate equal to said product of said predetermined constant upconst and said prior noise estimate,

if said current smoothed power estimate is less than a product of a predetermined constant downconst and said prior noise estimate, then setting a current noise estimate equal to said product of said predetermined constant downconst and said prior noise estimate, and

else setting a current noise estimate equal to said current smoothed power estimate.

18. (New) The method of claim 17, wherein:

said step of calculating a noise estimate further includes setting said predetermined constant upconst to limit increase in said noise estimate to less than 3 Db per second.

19. (New) The method of claim 17, wherein:

said step of calculating a noise estimate further includes setting said predetermined constant downconst to limit decrease in said noise estimate to less than 12 Db per second.

20. (New) The system of claim 9, wherein:

said noise suppression circuit operates to calculate a noise estimate by being operable to

set a current noise estimate equal to said product of said predetermined constant upconst and said prior noise estimate if a current smoothed power estimate is greater than a product of a predetermined constant upconst and a prior noise estimate,

set a current noise estimate equal to said product of said predetermined constant downconst and said prior noise estimate if said current smoothed power estimate is less than a product of a predetermined constant downconst and said prior noise estimate, and

else set a current noise estimate equal to said current smoothed power estimate.

21. (New) The system of claim 20, wherein:

said noise suppression circuit operates to calculate a noise estimate by being further operable to set said predetermined constant upconst to limit increase in said noise estimate to less than 3 Db per second.

22. (New) The system of claim 20, wherein:

said noise suppression circuit operates to calculate a noise estimate by being further operable to set said predetermined constant downconst to limit decrease in said noise estimate to less than 12 Db per second.